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Computationally efficient algorithm for high sampling-frequency operation of active noise control

Nirmal Kumar Rout[a,*], Debi Prasad Das[b], Ganapati Panda[c]

[a] School of Electronics Engineering, KIIT University, Bhubaneswar, India
[b] Process Engineering and Instrumentation Cell, CSIR-Institute of Minerals and Materials Technology, Bhubaneswar, India
[c] School of Electrical Sciences, Indian Institute of Technology, Bhubaneswar, India

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ABSTRACT

In high sampling-frequency operation of active noise control (ANC) system the length of the secondary path estimate and the ANC filter are very long. This increases the computational complexity of the conventional filtered-x least mean square (FXLMS) algorithm. To reduce the computational complexity of long order ANC system using FXLMS algorithm, frequency domain block ANC algorithms have been proposed in past. These full block frequency domain ANC algorithms are associated with some disadvantages such as large block delay, quantization error due to computation of large size transforms and implementation difficulties in existing low-end DSP hardware. To overcome these shortcomings, the partitioned block ANC algorithm is newly proposed where the long length filters in ANC are divided into a number of equal partitions and suitably assembled to perform the FXLMS algorithm in the frequency domain. The complexity of this proposed frequency domain partitioned block FXLMS (FPBFXLMS) algorithm is quite reduced compared to the conventional FXLMS algorithm. It is further reduced by merging one fast Fourier transform (FFT)-inverse fast Fourier transform (IFFT) combination to derive the reduced structure FPBFXLMS (RFPBFXLMS) algorithm. Computational complexity analysis for different orders of filter and partition size are presented. Systematic computer simulations are carried out for both the proposed partitioned block ANC algorithms to show its accuracy compared to the time domain FXLMS algorithm.

1. Introduction

Increased awareness of noise pollution has made the control of noise an important topic of research. There are two distinct approaches to control the acoustic noise: passive and active. The traditional approach to acoustic noise control employs passive techniques such as enclosures, barriers, and silencers. These passive silencers are effective for their high attenuation over a broad frequency range. Since the thickness of passive silencers is comparable to the wavelength of the noise signal it is impracticable to employ them at low frequency. To alleviate this difficulty the active noise control (ANC) has been introduced as a tool to control especially the low frequency noise up to 1 KHz. The ANC is an electroacoustic device based on the principle of destructive interference where the unwanted sound is canceled by generating an antisound (antinoise) of equal amplitude and opposite phase. The original unwanted sound and the antinoise acoustically combine in

* Corresponding author.
E-mail addresses: routnirmal@rediffmail.com (N.K. Rout), dpdas@immt.res.in, debi_das_debi@yahoo.com (D.P. Das), ganapati.panda@gmail.com (G. Panda).

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the acoustic medium, resulting in the cancellation of both sounds. The ANC uses the most conventional filtered-x least mean square (FXLMS) algorithm as its control algorithm due to its simplicity [1–3]. Least mean square (LMS) based adaptive filters are generally used for system identification, acoustic echo cancellation etc. The block diagrams of the LMS and FXLMS algorithms are presented in Fig. 1(a) and (b) respectively.

In the system identification problem, the output of adaptive filter $W(z)$ is compared with the output of the unknown system to generate the error. The error is used to update the adaptive filter. Whereas, in ANC, the output of the adaptive filter is passed through an electro acoustic path (called secondary path). The output of the secondary path is compared with the output of the primary path (in Fig. 1(b)) to get the error which is used to tune the adaptive filter. Since, the output of the adaptive filter is passed through the secondary path, a model of the secondary path is first estimated (called as secondary path estimate filter $\hat{S}(z)$) and the reference signal $x(n)$ is filtered through $\hat{S}(z)$ to tune the adaptive filter of the ANC. In a practical situation, the order of the secondary path estimate filter is very large and increases with increase in sampling-frequency. Increased sampling frequency is required to control higher frequency noise components. For example to control a noise signal of 1 kHz frequency, about 10 kHz sampling is required which necessitates higher order secondary path estimate filter and associated higher computational complexity and cost. Fig. 2 illustrates how the order of secondary path filter increases with the increase in the sampling frequency. As a typical example, an 8 kHz sampling rate requires about 300 filter coefficients to estimate the secondary path where as a 1 kHz sampling requires about 50 filter coefficient (as seen in Fig. 2).

Therefore, the conventional FXLMS algorithm becomes computationally expensive when the secondary path estimate as well as the controller use large number of filter coefficients. Many attempts have been made in the past to reduce such large computation. It is reported that fast implementation of active noise control in time domain [4–6] achieve about 25% computational saving compared to that of the conventional algorithms.

Researchers have utilized the frequency domain implementation of convolution and correlation operations using overlap-save method to develop frequency domain block LMS (FBLMS) algorithm which offers substantial computational savings over the time domain LMS algorithm for higher order adaptive filters [7]. The unconstrained frequency domain adaptive filter suggested in [7,8] has been reported as a useful alternative to the FBLMS algorithm which approximately implement linear convolution and correlation operations. Many applications are seen using frequency domain implementations of LMS algorithms such as optical communication [9], echo cancellations [10,11]. Convergence analysis of the FBLMS algorithm for finding the step-size bound is proposed recently in [12].

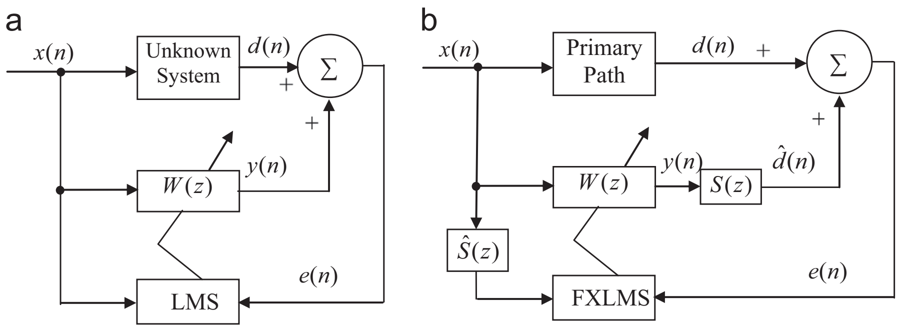

Fig. 1. Block diagrams: (a) LMS based system identification (b) FXLMS based ANC.

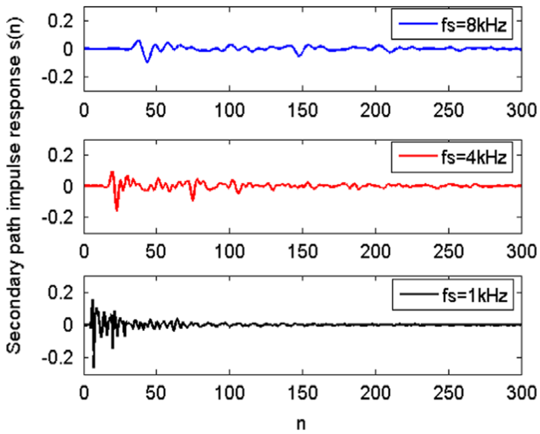

Fig. 2. Secondary path estimate impulse response by varying the sampling frequency.